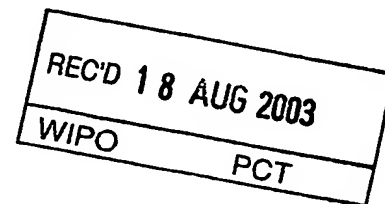**Europäisches Patentamt**

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Patentanmeldung Nr.   Patent application No. Demande de brevet n°

02077952.6

Der Präsident des Europäischen Patentamts;
Im Auftrag

For the President of the European Patent Office

Le Président de l'Office européen des brevets
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Bezeichnung der Erfindung/Title of the invention/Titre de l'invention:
(Falls die Bezeichnung der Erfindung nicht angegeben ist, siehe Beschreibung.
If no title is shown please refer to the description.
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Transmission of MPEG2 transport stream packets over IP

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Transmission of MPEG2 transport stream packets over IP

Why is it important now.
Transmission of data packets over the Internet is applied very often. However the Internet is a packet switched network, which means that every packet is transmitted independently from the other packets, there is in general no guarantee that the IP packets arrive after a
5   certain time. Every packet will have a different delay, even the order of the packets at the receiver might have been changed.
For Real-Time applications this is a problem, here the order of the packets should be kept the same and all packets should have, within a certain tolerance, the same overall delay.
An example of a Real-Time application is Transmission of MPEG encoded Audio and
10   Video (e.g. Video-On-Demand) signals. The Real-Time applications are getting more and more important.

Existing situation:
There has been standardized for the WWW a Real-Time Protocol (RFC1889). In
15   RFC1889 there are two different data packets:
   ▪ RTP: A Transport Protocol for Real-Time Applications. In these packets the data packets which carry the real-time properties are transmitted.
   ▪ RTCP: RTP Control Protocol, which is used to monitor the quality of service and to convey information about the participants in an on-going session. In the RTCP it is
20   possible to transmit both the RTP timestamp and the NPT. The NPT is the wall clock (the same all over the world). It is optional to implement an NPT at the receiver
However RFC1889 cannot be used on its own, an additional protocol is needed for a certain application. For the transmission of MPEG data, the additional protocol
(RFC2250) must be applied. Standardization of RFC2250 is not yet (completely) finished.
25

Problems with the existing situation.
   ❑ In RFC1889 a header is added to every IP packet. The header contains a time-stamp which is derived from the clock used in the application. The contents of the timestamp is
30   specified in RFC2250 (or any other application protocol if another application is transmitted). In RFC2250 the timestamp is derived from the 27 MHz clock used in
   MPEG. The resolution is ~11 msec (90 kHz clock) which is not sufficient for MPEG
   Transport streams. The timestamp with the higher resolution is in the TS packet itself (the PCR packet).
35   ❑ Also in an IP transmission this 27 MHz clock frequency (and phase) cannot be transmitted, the receiving clock should be locked with a PLL to the transmitter clock.
   This is very difficult if the jitter on the packets is very large.
   ❑ If the transmission delay is not known then the buffer size needed to compensate the jitter introduced by the network, is doubled. The jitter on the Internet can be large so the
40   size of the buffer will be large. Also the overall delay is larger than needed (up to two times).
   ❑ To increase the efficiency of the packing, a number of TS packets is packed in one IP packet. There is only one RFC1889 timestamp for the whole IP packet. This timestamp

refers to the first TS packet of the IP packet. The jitter on the applications packets which are packed in one IP packet is still remarkable (too high for MPEG). Jitter compensation for every individual TS packet is needed.

❑ The MPEG2 Transport Stream is not the only Real-time application. Other applications are e.g.:

- Transmitting DV packets over IP
- Transmitting IEC958 audio packets over IP
- Transmitting DSS packets over IP.

For these applications there is protocol yet.

Proposal:

❑ Using a Timestamp in the RFC1889 which is derived from a clock known in both the transmitter and the receiver. Then the transmission delay is known on receiving side. In this way the receiver buffer size (and delay) can be minimized.

❑ Packing of the MPEG TS packets in such a way in the IP packet that the remaining jitter on the TS packets can be compensated.

In the following some embodiments of the proposal are given.

Embodiment 1: Using modified RFC1889 protocol.

Transmit in the RTP packet header the NPT timestamp (see figure 1). The RTP timestamp is not derived from the clock used in the application but from the wall clock in the transmitter. The RTP time stamp represents the NPT value (at the transmitter) at the moment the first byte of the IP packet is delivered to the transmitter.

The wall clocks at the Transmitter and at the Receiver do have the same counter value. The delay from the RTP packet can be calculated by comparing the NPT time stamp with the NPT value of the receiver wall clock. A maximum overall delay should be realized. The delay during transmission is known, the remaining required delay in the receiver buffer can be calculated (overall delay minus delay during transmission). Once the delay of the first packet is set, then the delay of the other packets can be calculated. The size of the buffer is equal to [maximum bit rate * maximum delay of the packets].

In figure 2 it is explained how the constant delay is made. A variable input rate is (with a Peak-rate equal to the maximum rate) expected. The constant overall delay consists of a contribution from the transmitter and a contribution realized in the receiving buffer. The delays cannot be negative (lines A-B-C do not cross each other).

If the wall clock is not implemented in the receiver then the NPT counter is set to the value of the first RTP timestamp (expecting minimum delay in the transmission). The first packet is removed from the buffer after a time which is equal to the maximum delay in the transmission. Once the delay of the first packet is set, then the delay of the other packets can be calculated. The buffer size in the receiver is should be at least twice the (maximum bit rate * maximum delay of the packets). Expecting a maximum bit rate of 2 Mbyte/sec and a maximum delay of 1 second then a buffer size of 2 Mbyte can be saved by implementing the NPT wall clock.

The compensation of the jitter from the MPEG TS packets in the IP packet is explained in embodiments 3 and 4.

The advantage of this system w.r.t. the current situation, are:

- The application layer is de-coupled from the transmission. The method can be used for other applications too (like DV, DSS, IEC958).

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- The buffer size and the delay can be minimized by implementing the wall clock at the receiver.

5   *Embodiment 2: Minimizing the buffer size while using the existing protocols.*
     Transmit in the RTP header a timestamp which is derived from the clock used in the
     application. For MPEG Transport Streams this is specified in RFC2250. Transmit in a
     RTCP packet an RTP timestamp which is derived from the clock of the application and
     an NPT timestamp derived from the transmitter wall clock. Both timestamps refer to the
10   same moment in time. If the receiver implements the NPT wall clock then the buffer size
     can be reduced. This is explained in figure 3.
     At the transmitter a number of MPEG TS packets are packed in one IP packet. The RTP
     timestamp is derived from the STC counter, the value of the timestamp corresponds to the
     arrival time of the first byte of the first TS packet in the IP packet.
15   At regular instances an RTCP packet is interleaved between the RTP packets. The RTCP
     packets contains an NTP timestamp and an STC timestamp. Both timestamps are derived
     at the same moment.
     On the receiving side The NPT timestamp is compared with the Wall clock from the
     receiver. The Transmission delay of the RTCP packet is known. Now the STC counter can
20   be set to the correct value (same value as in the transmitter). The STC counter and the
     RTP timestamps are used to compensate the jitter of the IP packets. A maximum overall
     delay is expected. The delay during transmission is known, the remaining delay in the
     buffer can be calculated (overall delay minus delay during transmission). The size of the
     buffer is equal to [maximum bit rate * maximum delay of the packets].
25   The advantages of this system are: Existing protocols can be used. It is a manufacturers
     option to implement the Wall clock at the receiver. If the wall clock is not implemented
     then the STC counter is set to the value of the first RTP timestamp (expecting minimum
     delay in the transmission).. The first packet is removed from the buffer after a time which
     is equal to the maximum delay in the transmission. The buffer size in the receiver is
30   should be at least twice the (maximum bit rate * maximum delay of the packets).
     The compensation of the jitter from the MPEG TS packets in the IP packet is explained in
     embodiment 3 and 4.
     A remaining disadvantage is that only the MPEG TS transmission is standardized. Real-
     time protocols like RFC2250 for MPEG are not yet available for other Real-time streams (
35   DV, DSS, IEC 958).

     Once the jitter is compensated from the IP packets then still the jitter from the individual
     packets in one IP packet should be compensated. For understanding the timing of
     Transport Stream packets from a 'Full" and a "Partial" Transport stream are given in
40   figure 4. Packing a number Transport stream packets is shown in Figure 5.

     Embodiment 3: Jitter compensation of the MPEG TS packets using an additional time
                    stamp.
45   If the maximum size of an IP packet is limited to 1500 bytes then a maximum of 7 TS
     packets can be packed in one IP packet. The IP packets are still aligned with the MPEG
     TS packets, although this is not needed in this proposal. The method applied for
     transmitting MPEG Transport streams over Digital Interfaces is applied. To every TS
     packet a time stamp is added which contains the arrival time of the TS packet . The time
50   stamp is derived from the STC clock (27 MHz) which is used in the MPEG2 TS. In this

way Source packets with a size of 192 bytes are created. Up to 7 Source packets are
packed in one IP packet. Some examples are given in Figure 5.
It is expected that with the embodiments 1 or 2 the jitter of the IP packets is removed.

5   In figure 6 the method for compensating the jitter on the individual packets is shown for
the situation where embodiment 1 is used to compensate the jitter on the IP packets. In
the buffer [B2] the jitter introduced by packing 7 (or less) Source packets in one IP
packet, is compensated. The size of buffer [B2] is small (1 or 2 IP packets).

10   In figure 7 the method for compensating the jitter on the individual packets is shown for
the situation where embodiment 2 is used to compensate the jitter on the IP packets. Now
there is needed only one buffer .

The advantage of this method (using an additional time stamp for every TS packet) is that
15   it is also simple for Partial TSs .

Embodiment 4: Jitter compensation of the MPEG TS packets. No additional time stamp is used.

20   Again the maximum size of an IP packet is limited to 1500 bytes, a maximum of 7 TS
packets can be packed in one IP packet. The IP packets are still aligned with the MPEG
TS packets, although this is not needed in this proposal.   Up to 7 Source packets are
packed in one IP packet. Some examples are given in Figure 5.
It is expected that with the embodiments 1 or 2 the jitter of the IP packets is removed.
25

In figure 8 the method for compensating the jitter on the individual packets is shown for
the situation where embodiment 1 is used to compensate the jitter on the IP packets. In
the Receiver the STC counter is locked to the PCR in the MPEG TS. For this purpose it is
best to have as less as possible jitter on the arrival time of the PCR. The PCR packet
30   should be the first of the packets in the IP packet.

In the buffer [B2] the jitter introduced by packing 7 (or less) TS packets in one IP packet,
is compensated. There is a difference in the implementation for the "full" TS and the
'partial' TS.
35   For restoring a Full TS, all the packets between two PCRs are stored in Buffer [B2]. Then
the packets are equally distributed over the interval between the two PCR values. For this
method the buffer [B2] needs to be quite large (a few hundred kByte).

For restoring a Partial TS the following is carried out.
40   The jitter on the PCR packets is removed in the normal way (using the timestamp of the
PCR packet). The contents of the Video, audio and systems buffers is simulated. Every
next packet is removed from the buffer [B2] as soon as it is allowed by the buffer
constraints.
Note:
45   = The order of the packets has not been changed. Only one packet at a time needs to
be considered.
= The resulting stream might have gaps just before the PCR packet, but this is
allowed for partial Transport streams.

50

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In figure 9 the method for compensating the jitter on the individual packets is shown for the situation where embodiment 2 is used to compensate the jitter on the IP packets. Now there is needed only one buffer .. There is a difference in the implementation for the "full' TS and the 'partial' TS.

5

For restoring a Full TS, all the packets between two PCRs are stored in Buffer [B1+2]. Then the packets are equally distributed over the interval between the two PCR values. For this method the buffer needs to be larger (a few hundred kByte).

10   For restoring a Partial TS the following is carried out.
The jitter on the PCR packets is removed in the normal way (using the timestamp of the PCR packet). The contents of the Video, audio and systems buffers is simulated. Every next packet is removed from the buffer [B2] as soon as it is allowed by the buffer constraints.

15   Note:
    = The order of the packets has not been changed. Only one packet at a time needs to be considered.
    = The resulting stream might have gaps just before the PCR packet, but this is allowed for partial Transport streams.

20

Claims:

[1] = A method to transmit in the RTP protocol a time stamp which is independent from the clock frequency. The time stamp is derived from the wall clock at the transmitter.

= The 32 bits from the 64 bit NPT counter are selected in such a way that the resolution is high enough for accurate jitter compensation, and the maximum interval is larger than the maximum jitter on the IP packets.

= A method to minimize the buffer size and the delay on receiver side while using this time stamp.

= A method to minimize the buffer size and the delay on receiver side if the wall clock is not present in the receiver.

[2] = Using the RTCP protocol in such a way that the buffer size and the delay in the receiver is minimized.

= Repeating transmission of the RTCP packets in such a way that differences in time base between NPT clock and STC clock can be compensated.

= Checking the long term average amount of data in the buffer in order to discover difference in time base between NPT clock and STC clock.

[3] = Inserting a time stamp which is derived from the clock used in the application before every application packet. Using the time stamp from every first application packet of an IP packet (after IP packet jitter compensation) as the reference signal for the PLL on the receiver side.

[4] = Packing the MPEG TS packets in IP packets in such a way that the PCR packets are always the first TS packet in the IP packet.

[5] = Restoring a Full MPEG TS by storing all packets between two PCR packets and equally distributing these packets on the time axis between the two PCR values on the time axis.

[6] = Restoring a partial MPEG TS by putting the PCR packets on the time axis indicated by the PCR value and by simulating the contents of decoder / transmitting buffers and putting the non-PCR packets in such a way and one-by-one on the time axis, that no violation of the buffer constraints occurs.
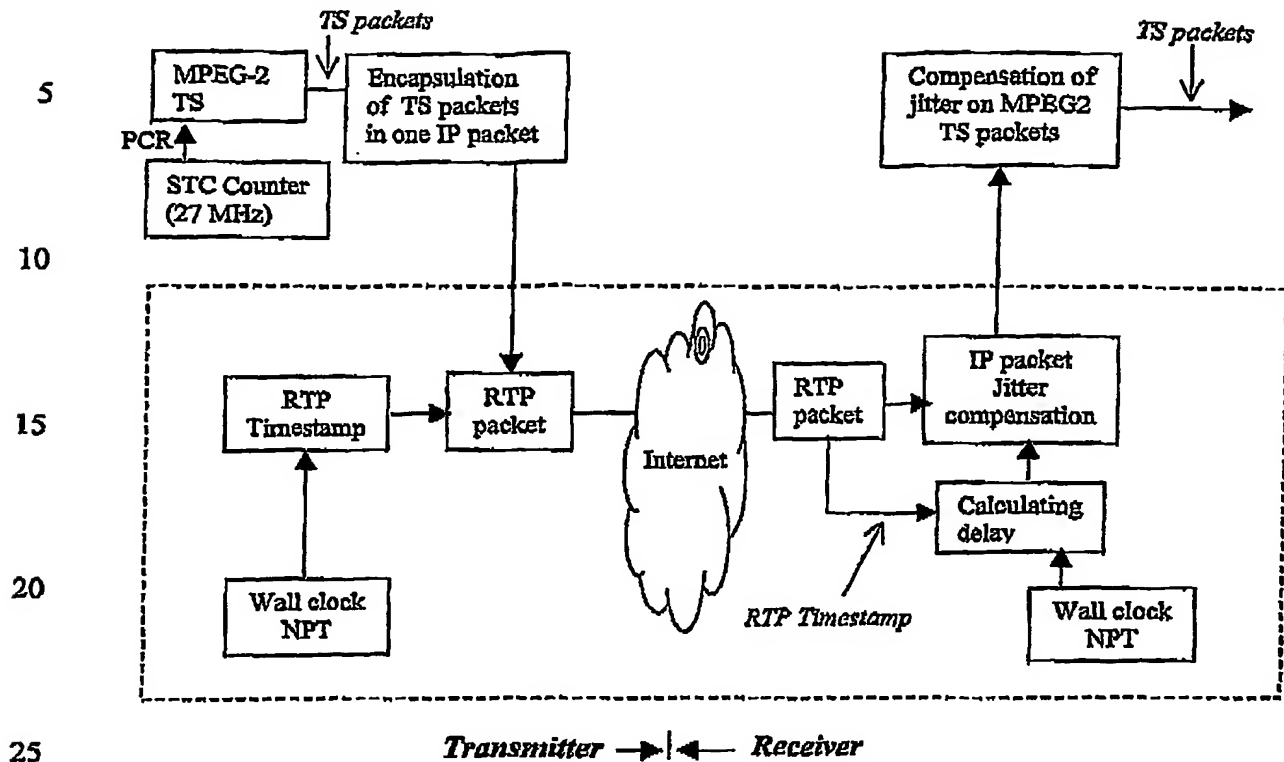
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TS packets

| MPEG-2 TS | Encapsulation of TS packets in one IP packet |

PCR

STC Counter (27 MHz)

Compensation of jitter on MPEG2 TS packets

TS packets

RTP Timestamp → RTP packet → Internet → RTP packet → IP packet Jitter compensation

Calculating delay

RTP Timestamp

Wall clock NPT

Wall clock NPT

IP packet Jitter compensation

Transmitter ➤ | ◄ *Receiver*

Figure-1: Block diagram of Transmitter / Receiver using modified RFC1889 .

STC = System Time Clock (time base of the MPEG coder), derived from 27 MHz clock

PCR = Program Clock Reference (resolution one period of the 27 MHz clock)

NPT counter contains 64 bits.
    b0..b31   for the fraction of a second.
    b32 .. b63 for the number of seconds

The RTP time stamp is 32 bits long . The accuracy is enough to realize a small IP packet jitter after Jitter compensation. This depends on the maximum jitter which can still be compensated in the jitter compensation of the MPEG packets. The maximum tolerance for MPEG packets is 500 nsec.
The Wrap-around in the 32 bit counter value should be larger than the maximum delay of the IP packets. Expecting a maximum delay of 1 second , then e.g.
RTP time stamp is b10 .. b41 of the NPT counter fulfils all requirements.
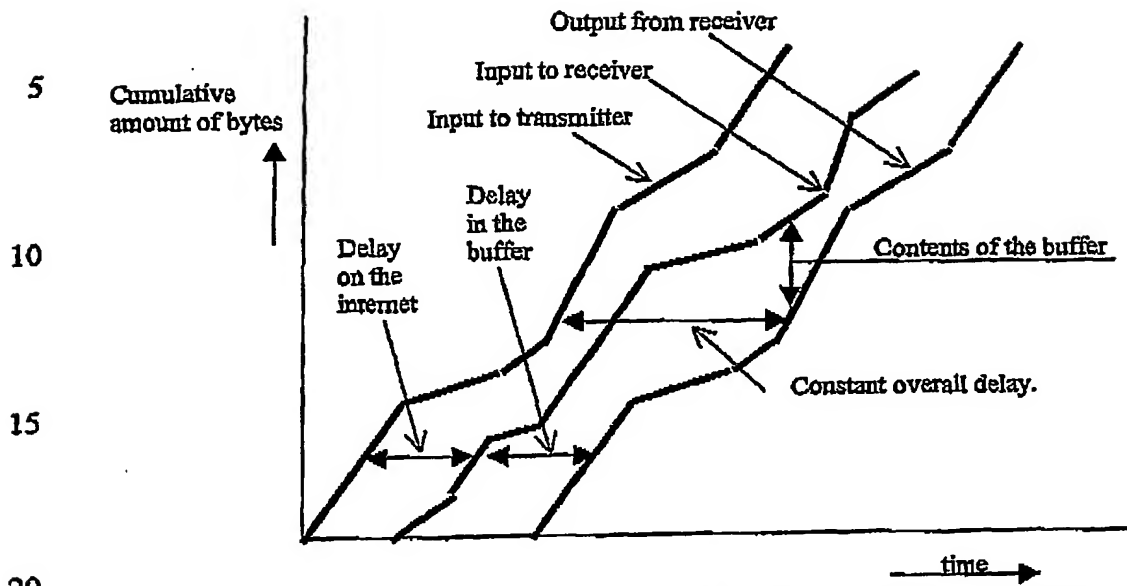
Cumulative
amount of bytes

Output from receiver

Input to receiver

Input to transmitter

Delay
in the
buffer

Delay
on the
internet

Contents of the buffer

Constant overall delay.

time

Figure 2: Cumulative amount of bytes as a function of time.
A = Input to the transmitter
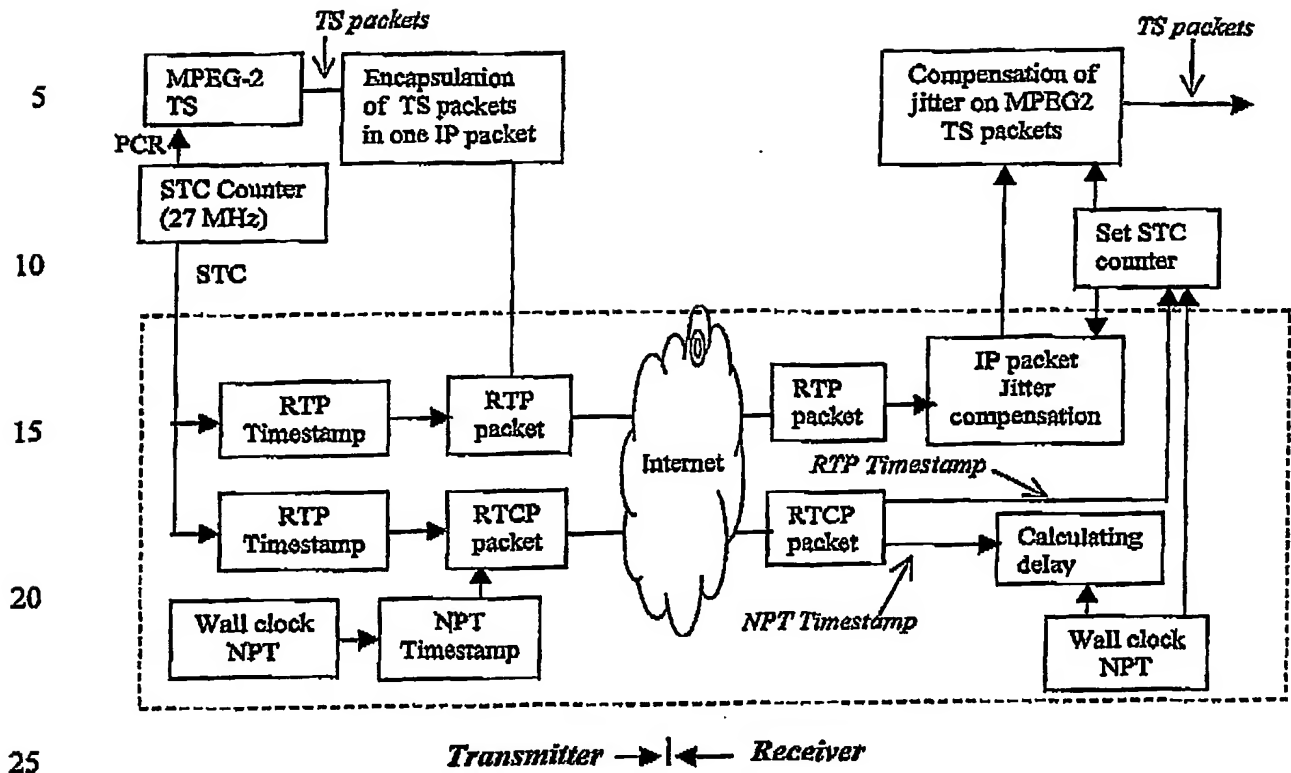B = Input to the receiver.
C = Output from the receiver.

5

10

15

20

25

Figure-3: Block diagram of Transmitter / Receiver using the RFC1889 with RTCP packets.

The RTP time stamp is 32 bits. The PCR is 42 bits (STC base + STC extension). So the range of the RTP time stamp is less then the range from the PCR. The RTP time stamp is derived from a 90 kHz clock, STC-base [RTP-timestamp is b9..b40 from the PCR].

The NPT timestamp in the RTCP packet is the full 64 bits and the RTP timestamp is again 32 bits (same as in RTP packet).

Note:
- The accuracy from the RTP timestamp should be large enough to have a small enough jitter from the IP packet (small enough to compensate jitter of the individual TS packets from the IP packet). The warp-around should represent a large enough time interval (larger than the maximum delay of the IP packet).
- The STC counter (on Record side) might drift away from the NPT counter. On the receiver side at regular instances RTCP packets must be used to check and if needed the STC counter value should be adjusted.
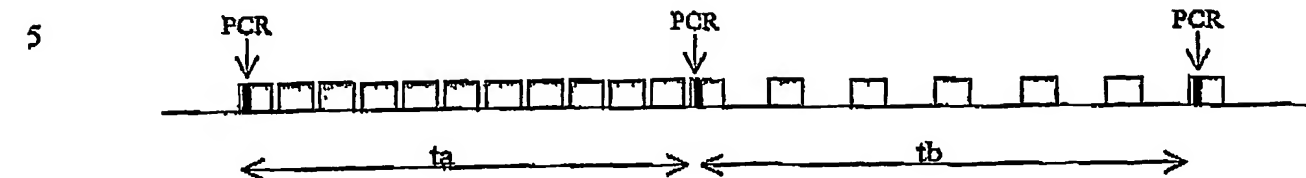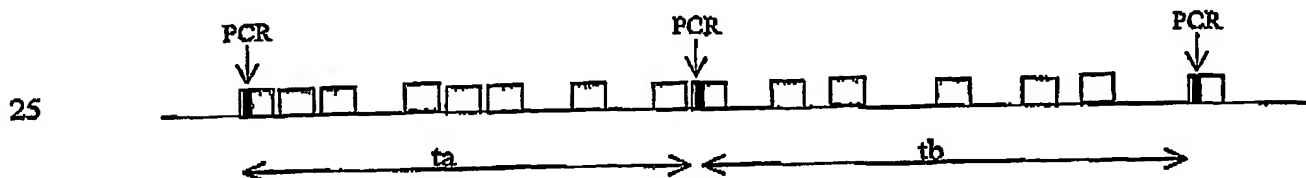
Figure 4a: MPEG TS packets from a full Transport Stream with variable bit rate.
In this example the Transport Stream contains only one PCR PID number.
In IEC13818-1 it is specified the bit rate between two PCR packets is
constant. Variable bit rate is realized by making a piece-wise constant bit
rate. The distance between PCR packets is < 100 msec (very often < 40
msec). With a bit rate of 2 Mbyte/second there are 200 kbytes between two
PCR packets if the distance is 100 msec and there are 80 kbytes if the
distance is 40 msec.

**Figure 4b: MPEG TS packets from a partial Transport Stream with variable bit
rate.**
In this example the Transport Stream contains only one PCR PID number.
In IEC61883-4 the partial Transport stream is explained. The bit rate need
not be constant between two PCR values. The buffer constraints from the
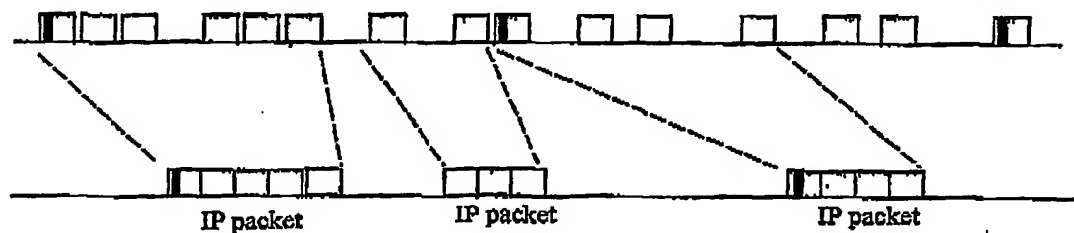STD model are still fulfilled.
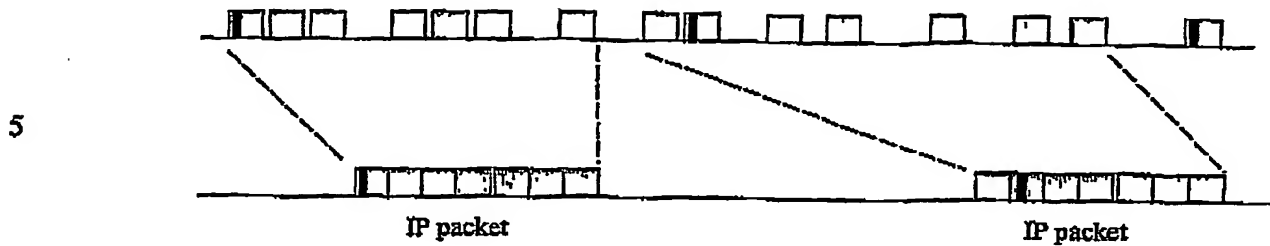
Figure 5a: Packing a number of MPEG TS packets in one IP packet.
There are no constraints location of the PCR packet in the IP packet.

Figure 5b: Packing a number of MPEG TS packets in one IP packet, PCR packets
are aligned.
If there is a PCR packet in the IP packet, then it is the first of the packets in the
IP packet.
The number of TS packets between two PCR packets is 200 .. 1000 for a 2
Mbyte/second bit rate. Efficiency is of packing reduced only a little if aligning
of PCR packets is applied.

5

188 byte
TS packets

MPEG2TS → Inserting Time stamp

192 byte
Source packets

10   PCR

STC PLL (27 MHz)   Creating IP packet payload

TS packet Jitter Compensation [B2] → TS packet 188 byte

Burst of Source packets

STC PLL [27 MHz]   PCR

15

RTP Timestamp → RTP packet → Internet → RTP packet → IP Jitter compensation [B1]

20   Calculating delay

Wall clock NPT   RTP Timestamp   Wall clock NPT

25

Transmitter →|← Receiver

30

Figure 6: Block diagram of Transmitter / Receiver using modified RFC1889

Buffer [B1] is used to compensate the jitter on the IP packets.
Buffer [B2] is used to compensate the jitter on the individual TS packets in a IP

35   packet.
In the Receiver the STC counter is locked to the PCR in the MPEG TS. For this
purpose it is best to have as less as possible jitter on the arrival time of the PCR.
The PCR packet should be the first of the packets in the IP packet.
The timestamp in front of the TS packet is used to compensate the jitter. The

40   size of the buffer [B2] is a few IP packets.
Note:
     After compensation of the IP packet jitter, the time stamp from the first TS
     packet of every IP packet can be used as input for the PLL. There are much
     more IP packets than PCR packets, so PLL is more stable.

45

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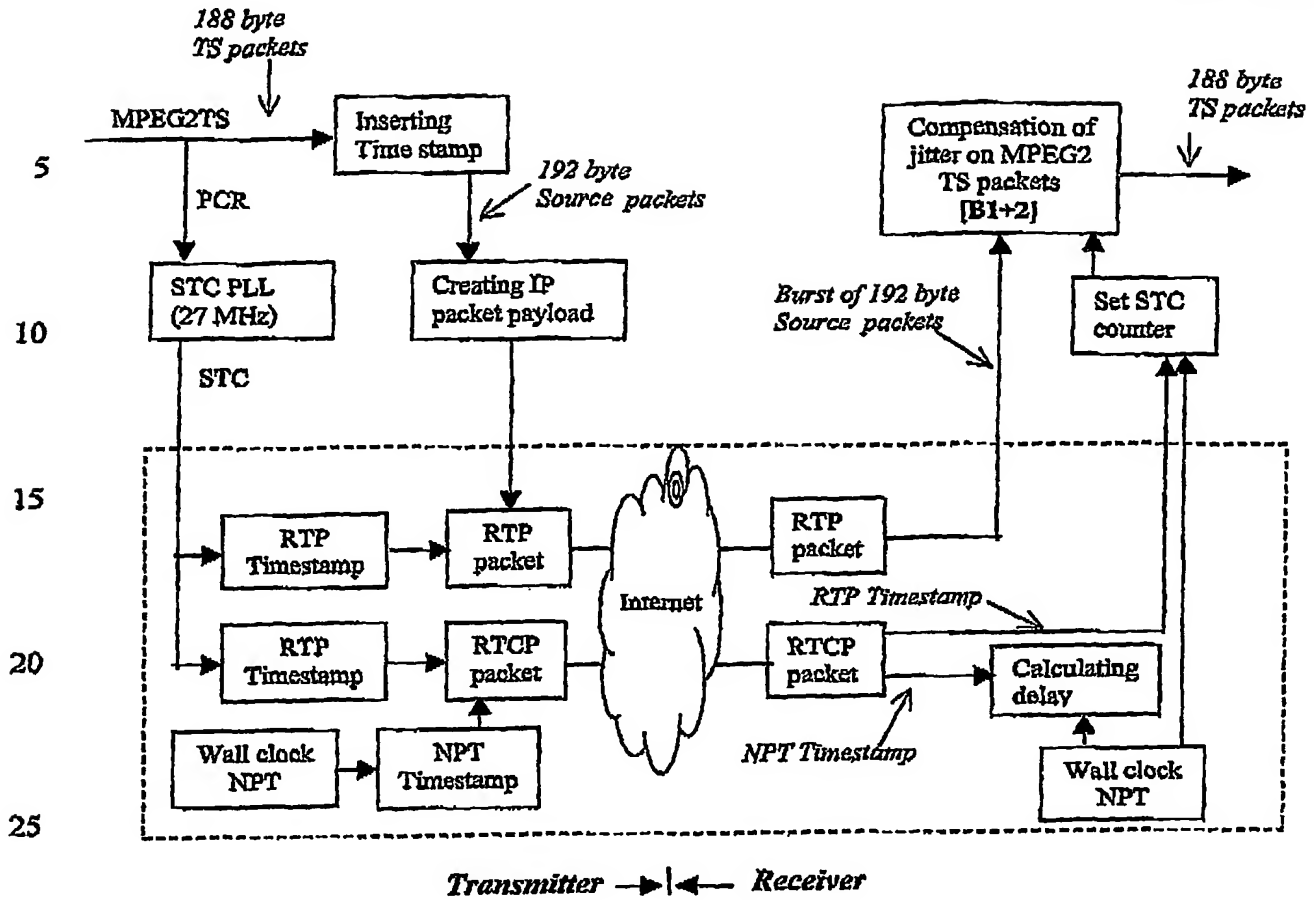
Figure-7: Block diagram of Transmitter / Receiver using the RFC1889 with RTCP packets.

On the Receiver there is no PLL.

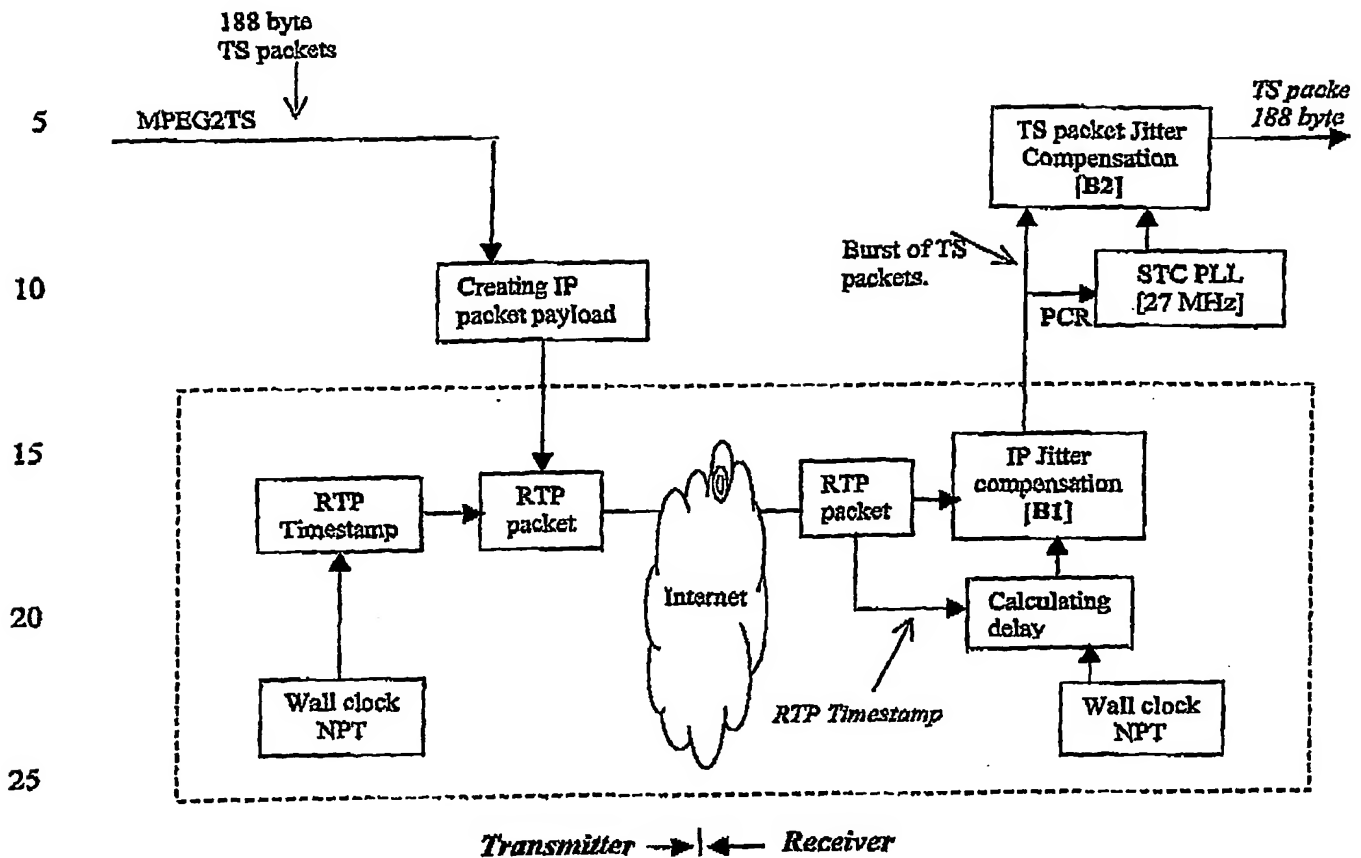The time stamp in front of every Ts packet is used to compensate the jitter of each TS packet.

Figure-8: Block diagram of Transmitter / Receiver using modified RFC1889.
Buffer [B1] is used to compensate the jitter on the IP packets.
Buffer [B2] is used to compensate the jitter on the individual TS packets in a IP
packet.
In the Receiver the STC counter is locked to the PCR in the MPEG TS. For this
purpose it is best to have as less as possible jitter on the arrival time of the PCR.
The PCR packet should be the first of the packets in the IP packet.

For restoring a Full TS, all the packets between two PCRs are stored in Buffer
[B2]. Then the packets are equally distributed over the interval between the two
PCR values. For this method the buffer [B2] needs to be quite large (a few
hundred kByte).

For restoring a Partial TS the following is carried out.
The jitter on the PCR packets is removed in the normal way (using the
timestamp of the PCR packet). The contents of the Video, audio and systems
buffers is simulated. Every next packet is removed from the buffer [B2] as soon
as it is allowed by the buffer constraints.
Note:= The order of the packets has not been changed. Only one packet at a
time needs to be considered.
    = The resulting stream might have gaps just before the PCR packet, but
        this is allowed for partial Transport streams.

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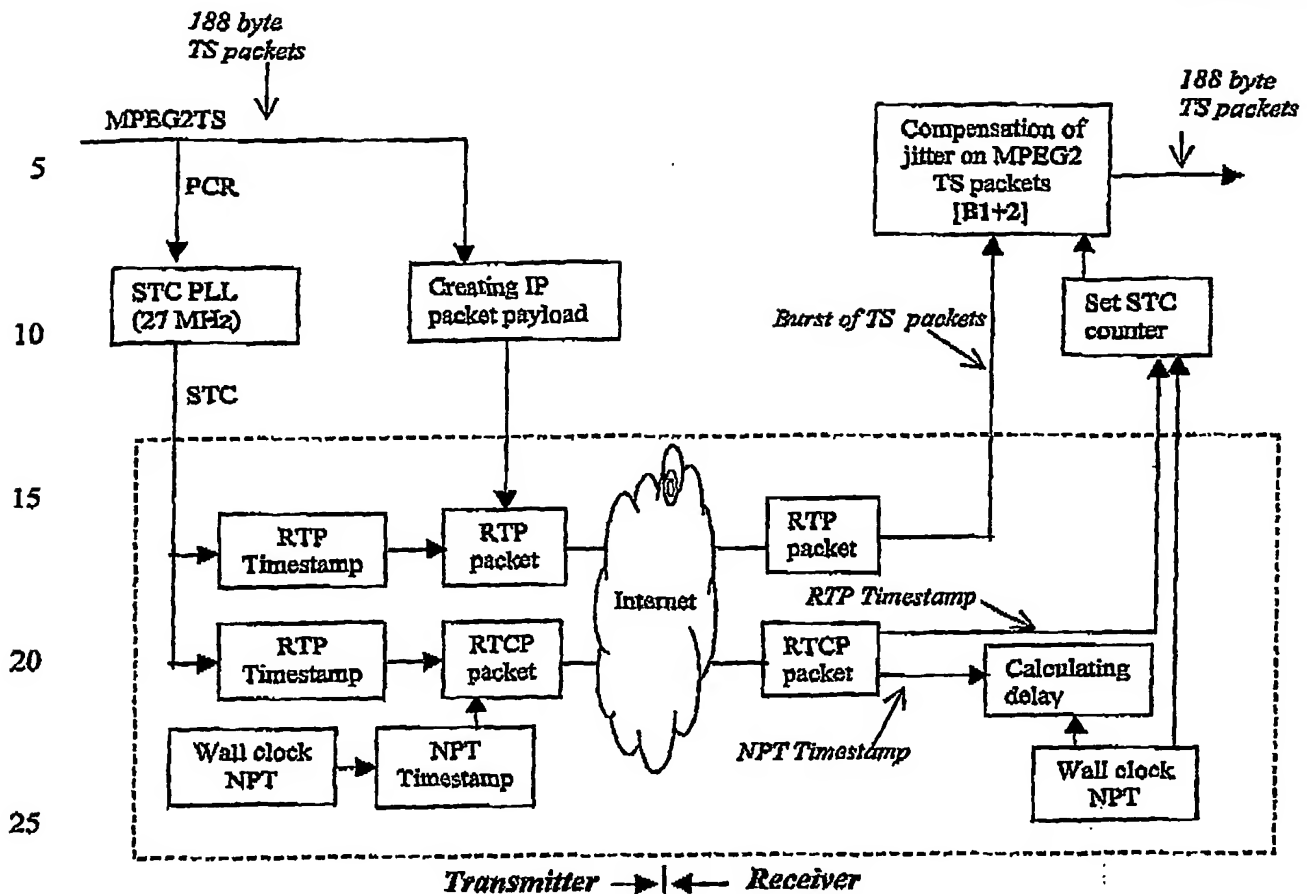15   19.07.2002

Figure-9: Block diagram of Transmitter / Receiver using the RFC1889 with RTCP packets.
On the Receiver there is no PLL, but RTCP packets are needed at regular distances (or the average content of the buffer should be kept constant).

For restoring a Full TS, all the packets between two PCRs are stored in Buffer [B1+2]. Then the packets are equally distributed over the interval between the two PCR values. For this method the buffer needs to be quite large (a few hundred kByte).

For restoring a Partial TS the following is carried out.
The jitter on the PCR packets is removed in the normal way (using the timestamp of the PCR packet). The contents of the Video, audio and systems buffers is simulated. Every next packet is removed from the buffer [B2] as soon as it is allowed by the buffer constraints.
Note:= The order of the packets has not been changed. Only one packet at a time needs to   be considered.
      = The resulting stream might have gaps just before the PCR packet, but this is allowed for partial Transport streams.